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(54) A HARMONIC AND FREQUENCY-LOCKED LOOP PITCH TRACKER AND SOUND SEPARATION SYSTEM

HARMONISCHE UND FREQUENZSTARRE GRUNDFREQUENZ-FOLGESCHALTUNG SOWIE SYSTEM ZUR TRENNUNG VON GERÄUSCHEN

DETECTEUR DE HAUTEUR DES SONS A BOUCLE VERROUILLEE EN HARMONIQUES ET EN FREQUENCE ET SYSTEME DE SEPARATION DES SONS

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(56) References cited:

US-A- 3 978 287	**US-A- 4 004 096**
US-A- 4 486 900	**US-A- 4 495 475**
US-A- 4 523 328	**US-A- 4 890 071**
US-A- 5 157 623	**US-A- 5 285 475**
US-A- 5 353 372	

- **PROCEEDINGS OF THE SEVENTH SSAP WORKSHOP, 26-29 June 1994, KUMARESAN et al., "RISC: An Improved Cosatas Estimator-Predictor Filter Bank for Decomposing Multicomponent Signals", pages 1-4.**
- **TWENTY-SEVENTH ASILOMAR CONFERENCE ON SIGNALS, SYSTEMS AND COMPUTERS, 1-3 November 1993, KUMARESAN et al., "On Separating Voiced-Speech Into Its Components", pages 1041-1046.**
- **PROCEEDINGS OF THE IEEE INTERNATIONAL CONFERENCE ON ACOUSTICS, SPEECH AND SIGNAL PROCESSING (ICASSP-94), April 1994, RAMALINGAM et al., "Voiced-Speech Analysis Based on the Residue Interfering Signal Canceler RISC Algorithm", pages I-473 - I-476.**

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Description

[0001] The present invention relates generally to pitch tracking systems, methods for tracking the pitch of a quasi periodic sound source and for the separation of periodic signals from mixtures of sounds.

BACKGROUND OF THE INVENTION

[0002] Pitch tracking is of interest whenever a single quasi periodic sound source is to be studied or modeled. For instance, the trajectory of a sound's pitch, also called the fundamental frequency, over a period of time can also be used to synthesize similar or related sounds using speech or musical synthesis techniques. An example of a quasi periodic sound source is a singer's voice singing a particular note (e.g., high C). The sound generated by the singer typically has a certain amount of vibrato or pitch modulation, noise and aperiodicity in the wave shape, making the sound quasi periodic rather than a pure periodic signal.

[0003] Currently pitch detection methods can be classified into three categories: Fourier-based frequency domain techniques, time domain techniques, and methods which use both techniques. The present invention is a time domain technique.

[0004] In time domain "feature detection methods", the input signal is usually preprocessed to accentuate some time domain feature, and the time between occurrences of that feature is calculated as the period of the signal. The pitch and the period of the input signal are related by the equation: pitch = 1/period. A typical time domain feature detector includes a low pass filter for detecting peaks or zero crossings of the filtered signal. Since the time between occurrences of a particular feature is used as the period estimate, feature detection schemes usually do not use all of the data available. Selection of a different feature often yields a different set of pitch estimates. Since estimates of the period are often defined at the instant when the features are detected, the frequency samples yielded are not uniformly distributed in time. To avoid the problem of non-uniform time sampling a window of fixed size can be moved through the signal in order to obtain an averaged period estimate.

[0005] Other prior art time domain methods include the use of auto correlation functions or difference norms to detect the similarity between the wave form and a time lag version of itself. However, prior art methods were computationally inefficient, with real time performance infeasible.

[0006] An example of a prior art method is disclosed in Kumaresan et. al. "RISC: An improved Costas Estimator-Predictor, Filter Bank for Decomposing Multicomponent Signals, Proc. 7th SSAP Workshop, 1994, pp 207-210.

SUMMARY OF THE INVENTION

[0007] According to the invention there are provided a frequency-locked loop pitch tracker as set out in claim 1 and a frequency-locked loop method for tracking an input signal as set out in claim 8.

[0008] In summary, the present invention is a system and method for tracking the pitch of a quasi periodic signal in a mixture of signals. The quasi periodic signal is "frequency warped" by selectively frequency modulating it, thereby resulting in a signal that is stationary and is a simplified spectrum which is more amenable to analysis. The resultant demodulated signal is low pass filtered resulting in an analytic signal whose phase winding rate is the frequency mismatch error between the target signal and the demodulating signal. The phase is differenced by multiplying the signal with a delayed version of itself creating an instantaneous autocorrelation. Thereafter the phase difference is measured with a complex arctangent to yield a resulting phase error. The resulting phase error is input to an integrator whose output value is the estimate of the frequency. This output frequency parameter is then used to update the demodulating signal thus closing the signal loop.

[0009] In a second embodiment of the present invention, a plurality of frequency locked loop trackers are servoed together centering each one of the trackers on a multiple of the fundamental frequency of the input signal. The resulting phase errors derived from the frequency lock loop trackers are weighted to improve system performance. In one embodiment, the frequency corrections from each tracker are weighted with the inverse variance of its tracking performance. Accordingly, harmonics with low variance are weighted strongly, and harmonics in a noisy region of the spectrum and thus high variance will be weighted less strongly. The resulting fundamental frequency estimate is a minimum-variance estimate, and is better than the best single frequency locked loop estimate. The weighted phase error is then fed back to an integrator to yield a high resolution estimate of the target signal fundamental frequency and all of its harmonics. The amplitude envelopes for each partial signal can be easily extracted and used in conjunction with the fundamental estimate from each frequency lock loop tracker to resynthesize the signal in isolation from the mixture. Since the resynthesized signal is in phase with the original signal, the target may be removed from the mixture by subtraction.

BRIEF DESCRIPTION OF THE DRAWINGS

[0010] Additional objects of interest to the invention will be more readily apparent from the following description and appended claims when taken in conjunction with the drawings, in which:

Figure 1 is a frequency locked loop tracker according to the preferred embodiment of the present invention.
Figure 2 shows the frequency locked loop tracker of Figure 1 including a phase locked loop.
Figure 3 shows the frequency locked loop tracker of Figure 1 including an improved frequency estimation means outside the tracking loop.
Figure 4 is a frequency locked loop tracker according to the preferred embodiment of the present invention including a resynthesis module.
Figure 5A shows the frequency locked loop tracker of Figure 4 including a delay line for compensating for the low pass filter group delay.
Figure 5B shows the frequency locked loop tracker of Figure 5A including a subtraction module for removing the resynthesized partial signal from the input signal.
Figure 6A is a frequency locked loop tracker according to Figure 3 including a resynthesis module.
Figure 6B shows the frequency locked loop tracker of Figure 6A including a subtraction module for removing the resynthesized partial signal from the input signal.
Figure 7 is a harmonic locked loop tracker in which a plurality of frequency locked loop trackers according to the preferred embodiment of the present invention are servoed for tracking a partial signal and a plurality harmonics of the partial signal.

DESCRIPTION OF THE PREFERRED EMBODIMENTS

[0011] Referring to Figure 1, the pitch tracker of the present invention 100 is shown. The pitch tracker 100 receives as an input signal *z*[n] 102 which is a mixture of a p[n] complex valued discrete time signal and some unknown disturbance signal *v*[n] wherein

$$z[n] = p[n] + v[n]$$

The target signal p[n] is a complex value discrete time signal defined for n>0 with a sampling frequency f_s wherein

$$p[n] = a[n]\exp\left(\frac{j2\pi}{f_s}\sum_{k=1}^{n} f[k] + j\Phi_0\right)$$

where a[n] is the instantaneous amplitude envelope,
f[n] is the instantaneous frequency, and
ϕ_0 is the phase offset at time n=0.

The first step in the analysis of the input signal z[n] 102 is to demodulate the input signal by means of a frequency matched demodulation signal. In particular, the input signal *z*[n] 102 is demodulated by multiplier 104, which multiplies the input signal z[n] with the complex conjugate of a frequency warping signal Ξ[n] 106. The use of the frequency warping signal 106 allows for the elimination of the FM band width component due to the instantaneous frequency modulation of the carrier. The frequency warping signal 106 demodulates the input signal *z*[n] 102 by means of a signal which is frequency matched to the input signal *z*[n] 102. In the preferred embodiment of the present invention, the input signal z[n] is demodulated using a complex phasor which rotates at a frequency equal to a frequency estimate generated by the pitch tracker 100. The frequency matching will be described in greater detail below in conjunction with the frequency estimate generated by the pitch tracker of the present invention. For the purposes of this first step of the analysis, it will be assumed that a frequency matched demodulation signal is provided. Those ordinarily skilled in the art will recognize that if the frequency estimate is equal to the target frequency, then the frequency matched demodulation by the instantaneous frequency *f*(t) of the estimate signal will yield a constant phase signal *d*[n] at or near DC.

[0012] The second step of the analysis requires low pass filtering of the constant phase signal to improve the signal to noise ratio. In particular, the complex demodulated signal d[n] resulting from the multiplication of the input signal z [n] 102 with the complex conjugate of the frequency warping signal 106 is coupled to a low pass filter 108. The low pass filter 108 improves the signal to noise ratio by low pass filtering the demodulated signal d[n] thereby attenuating the demodulated noise portion of the input signal.

[0013] In the preferred embodiment of the present invention, the low pass filter has a cut off frequency of f_c and unity gain at DC. The low pass filter may be of time-varying or time-invariant form with a fixed f_c. A time-varying filter can be used with a dynamically adjustable bandwidth wherein a wide cut-off frequency is programmed before frequency lock is achieved, and thereafter bandwidth can be reduced. However, dynamically altering the filter characteristics may introduce artifacts into the filter output if changes are made suddenly. Accordingly, in the preferred embodiment of the present invention, a time-invariant filter with a wide bandwidth is utilized providing a wide frequency lock-in range. A typical cut-off frequency would be 50-100 Hz. Wider cut-off frequencies are beneficial for tracking signals with rapidly varying frequency modulation, whereas narrower cut-off frequencies allow for better noise rejection.

[0014] In the next step of the analysis, the resultant low pass filtered signal is sampled to measure the phase difference of the filtered signal. The resultant signal u[n] is multiplied by means of multiplier 110 with a delayed and complex-conjugated version of itself via delay line 112. The change in phase of the resultant signal u[n] from the low pass filter 108 is then calculated by using a standard argument function 114 in order to result in the change in phase $\Delta\phi_u[n]$.

[0015] The frequency tracking error at time [n] is thereafter defined as $\varepsilon_f[n]$ where

$$\varepsilon_f[n] \equiv \frac{f_s}{2\pi}\Delta\phi_u[n]$$

Accordingly the change in phase $\Delta\phi_u[n]$ is normalized by multiplying the change in phase signal by the sampling frequency divided by 2π ($f_s/2\pi$) by multiplier 116 and results in an instantaneous frequency tracking error at time [n]. Note that the scaling factor may be left off resulting in calculations in radians per sample as opposed to hertz. In the preferred embodiment of the present invention the sampling frequency is 44,100 Hz, however, other sampling frequencies as is known in the art may be utilized. The frequency tracking error represents the error between the frequency estimate (generated by the pitch tracker 100 for use in demodulating the input signal z[n]) and the frequency of the target signal p[n].

[0016] Having calculated the frequency tracking error, the pitch tracker 100 utilizes this error information to generate a better frequency estimate for use in demodulating the input signal. Specifically, the frequency tracking error $\varepsilon_f[n]$ is combined with an attenuation tracking gain signal g[n] by multiplier 118 for input into integrator 120. The gain signal g [n] controls how fast the system will adapt to the particular frequency error $\varepsilon_f[n]$. The combination of the frequency error $\varepsilon_f[n]$ and the gain signal g[n] yields an attenuated frequency error signal. The attenuated frequency error signal is coupled to an integrator 120 in order to derive the estimated frequency output $\hat{f}[n]$ for use in updating the demodulation signal. Those ordinarily skilled in the art will recognize that any filtering or smoothing means may be used as is known in the art in lieu of the simple attenuated frequency integrator. In the preferred embodiment the integrator output, which reflects the estimated frequency of the target signal, must be initialized for tracking a particular desired partial signal. This may be accomplished by providing a particularized user input associated with the frequency of a particular partial signal to be tracked or may be accomplished by performing a sweep over an audio band in order to isolate a particular partial signal. Alternatively, a peak-detection scheme may be used on a FFT of an initial segment of the input signal to find a candidate initial frequency. Those ordinarily skilled in the art will recognize that the frequency tracker 100 will naturally track the strongest sinusoidal in the pass band of the low pass filter, and accordingly, the accuracy of the initial frequency estimate is not critical.

[0017] Finally the loop is closed by providing the frequency estimate to a phase accumulator for updating the frequency warping signal for use in demodulating the input signal. Specifically, the integrator estimated frequency output $\hat{f}[n]$ from integrator 120 is scaled via multiplier 122 by combining the estimated frequency with a scaling signal ($2\pi/f_s$ where f_s is the sampling frequency). The scaled output is coupled to a phase accumulator 124 for use in deriving an estimated phase responsive to the estimated frequency $\hat{f}[n]$. The estimated phase is then used as the estimated phase of the demodulating phasor to produce the warping signal 106 for use in the demodulation of the input signal $z[n]$. The phase accumulator 124 includes an integrator which derives an estimated phase from the scaled estimated frequency provided from the integrator 120. The derived phase is the estimated phase of the demodulating phasor for use in demodulating the input signal $z[n]$. In the preferred embodiment, this is accomplished by transforming the estimated phase into a sinusoid by taking the cosine and sine of the phase to generate a complex sinusoidal signal. Additionally, the phase is wrapped in a periodic fashion in order to prevent overflow of the phase accumulator 124.

[0018] Those ordinarily skilled in the art will recognize that the combination of the output estimate frequency from the integrator 120 in conjunction with the scaling multiplier 122 and the modulator 124 for deriving a frequency warping signal 106 is equivalent to a voltage controlled oscillator wherein the input frequency is used to derive a frequency matched demodulation signal. As such, the description of the integrator and phase accumulator according to the preferred embodiment should not be construed as limiting.

[0019] Referring now to Figure 2, the frequency locked loop tracker of the present invention is shown including a phase-locked loop for more feedback control. In this embodiment, a phase-locked loop is provided for locking to the phase of the demodulated and filtered signal *u*[n] described in conjunction with the first embodiment above. In the preferred embodiment described above, the frequency of a target signal is tracked but the phase is not. By providing a phase-lock feedback term, phase lock as well as frequency lock may be attained. The extra phase information provides for better isolation of the target signal for subtractive analysis. In this embodiment, the pitch tracker is more sensitive to noise and phase locking is difficult to attain in rapidly changing signals. Again, the analysis begins by demodulating a complex input signal *z*[n] 102 via multiplier 104 by a frequency warping signal 106 resulting in the complex demodulated signal d[n]. The complex demodulated signal d[n] is coupled to a low pass filter 108 producing an analytic output u[n].

[0020] The analytic signal u[n] is used in achieving phase lock by adding a modification to the frequency lock method described in the preferred embodiment. The phase lock loop is created by providing a second loop for tracking the phase mismatch error between the frequency warping signal 106 and the input signal *z*[n] 102. This is accomplished by taking the argument 202 of the analytic signal u[n] which yields a phase error. The resultant phase error is attenuated by a phase gain signal $g_\phi[n]$ via multiplier 204. The resultant attenuated phase error signal is coupled to the phase accumulator 124 of the preferred embodiment. Internal to the phase accumulator 124, this attenuated phase error is combined via an internal integrator with the derived phase estimate for phase lock. Those ordinarily skilled in the art will recognize that there are now two competing forces trying to guide the tracking. Close attention must be paid to the relative ratios of the gain g_n and the phase gain $g_\phi[n]$ since both phases range over $[-\pi, \pi]$. Accordingly, g[n] must be much greater than $g_\phi[n]$. However, as frequency lock is obtained, the phase gain $g_\phi[n]$ can be varied to be large enough to ensure that quick phase tracking convergence occur. Those ordinarily skilled in the art will recognize that automatic gain control algorithms which track the status of the frequency lock can adjust the gain g[n] and phase gain $g_\phi[n]$ making them dependant on the variances in the phase difference $\Delta\phi_u[n]$ and the phase mismatch error ϕ_u.

[0021] Referring now to Figure 3, the present invention is shown including a second frequency estimate $\hat{f}^{\dagger}[n-\delta1-\delta2]$ for providing a frequency estimate including group delay compensation outside the "loop" for use in resynthesis or other means as is known in the art. The basic tracking loop is identical to that shown in Figure 1, however, a second frequency estimate is made outside of the loop based on the crude estimates of $\hat{f}[n]$ from a first pass of a partial signal to be tracked along with the error estimation updates $\varepsilon_f[n]$, The crude estimates are then refined using a Kay optimal phase-difference smoother.

[0022] Specifically, the estimated frequency $\hat{f}[n]$ output from the integrator 120 is coupled via a delay line 304 to the frequency error signal $\varepsilon_f[n]$ via adder 306. Since the new estimate is made outside the loop, the new estimate does not contribute to tracking dynamics. The group delay of the low pass filter 108 is taken into account by the delay line 304. The output of the adder 306, which is effectively the phase difference of the input signal if it had not been demodulated by the frequency warping signal 106, is then coupled to a Kay smoother 302 having a group delay of δ2. In the preferred embodiment, the Kay smoother 302 is simply an FIR filter with quadratic coefficients given by the formula

$$w_{kay}[n] = \frac{6N}{N^2-1}\left\{\frac{n}{N} - \left(\frac{n}{N}\right)^2\right\}$$

for $1 < n \leq N-1$.

The Kay smoother output then reflects an improved estimate of the frequency being tracked. This improved estimate $\hat{f}^{\dagger}[n-\delta1-\delta2]$ may be used in providing a resynthesized partial signal as will be described below.

[0023] Referring now to Figure 4, the frequency locked loop tracker 100 of the preferred embodiment of the present invention is shown including a resynthesis module 401. Often it may be desired to produce a resynthesized partial signal p[n] which is a cleaned up version of the partial signal p[n] being tracked from the input signal *z*[n]. The cleaned up signal may be derived by combining the frequency warping signal 106 with the analytic signal u[n] via multiplier 402. The resultant output of this combination is an estimated partial signal p[n] which reflects the combination of the estimated frequency from the integrator 120 (as embodied in the frequency warping signal 106) combined with the envelope signal u[n].

[0024] Those ordinarily skilled in the art will recognize that this frequency locked loop tracker does not compensate for the group delay of the low pass filter 108. A better estimation of the partial signal $p[n-\delta_1]$ can be derived by providing a delay line 502 as shown in Figure 5A. The delay line 502 provides compensation for the group delay of the low pass filter and accordingly provides a more accurate resynthesized partial signal. Specifically, the delay line 502 couples the frequency warping signal 106 to the multiplier 402 yielding an improved estimate that accounts for the group delay of the low pass filter.

[0025] In addition to the isolation of a particular partial signal from a given input signal as described above, it is often desirous to produce a filtered input signal which has had the target signal removed. Examples of applications where this may be used is in the removal of a "voice" or musical instrument from a musical selection (e.g. audio signal) or the removal of background noise from a "voice". This process is known as notch-filtering, and when applied will result in a notch-filtered output signal. In the preferred embodiment, the partial signal p[n] or p[n-δ_1] may be used in a notch-filter process to derive a notch-filtered output signal as shown in Figure 5B. The notch-filtered output signal is derived by subtracting the resynthesized partial signal p[n] from the input signal z[n]. In the preferred embodiment, the input signal z[n] is coupled via a second delay line 504 to a first input of a subtractor 506. The second input of the subtractor 506 receives the resynthesized partial signal p[n-δ_1] from above. The subtractor 506 outputs a notch-filtered signal resulting from the subtraction of the partial signal from the input signal.

[0026] Referring now to Figure 6A, a second resynthesis module 601 for resynthesizing a partial signal is shown. The basic frequency locked loop tracker of Figure 1 is included with the Kay smoother filter of Figure 3 in order to make use of the improved frequency estimate $f^\dagger$[n-δ_1-δ_2] in producing a resynthesized partial signal. Specifically, the improved frequency estimate $f^\dagger$[n-δ_1-δ_2] is scaled by combining it with a scaling signal ($2\pi/f_s$ where f_s is the sampling frequency) via multiplier 604. The scaled frequency is then coupled to a second phase accumulator 602 which integrates the scaled frequency to create an improved estimated phase of the demodulating phasor for the phase accumulator 602. The phase accumulator 602 outputs a second frequency warping signal 606 which is utilized in demodulating a delayed version of the input signal z_n. This is accomplished by coupling the input signal z_n via delay line 608 to multiplier 610 for combining with the second frequency warping signal 606.

[0027] The complex demodulated signal $d^\dagger$[n-δ_1-δ_2] is then coupled to a second low pass filter 612 having a group delay of δ3. The output of the second low pass filter 612 is coupled with the second frequency warping signal 606 via multiplier 614 in order to yield an improved partial signal $p^\dagger$[n-δ1-δ2-δ3]. The second low pass filter is the resynthesis filter, and is designed to allow for higher-quality filtering characterized by a narrower cut-off frequency and linear phase response. Those ordinarily skilled in the art will recognize that a delay line 616 may be used to couple the second frequency warping signal 606 to the multiplier 614 in order to account for the group delay of the second low pass filter 612. Accordingly, the resultant output of the combination of the delayed second frequency warping signal 606 and the analytic signal from the low pass filter 612 will result in an improved partial signal $p^\dagger$[n-δ_1-δ_2-δ_3]. Because this resynthesized signal is generated outside the normal tracking loop, no tracking dynamics will be affected by this resynthesis function. Those ordinarily skilled in the art will recognize that the more efficient estimate of the partial signal p[n] can be used to calculate a high quality notched filter signal as is known in the art.

[0028] Again, the partial signal p[n-δ_1-δ_2-δ_3] may be used in a notch-filter process to derive a notch-filtered output signal as shown in Figure 6B. The notch-filtered output signal is derived by subtracting the resynthesized partial signal p[n] from the input signal z[n]. In the preferred embodiment, the input signal z[n] is coupled via a fourth delay line 618 to a first input of a subtractor 620. The second input of the subtractor 620 receives the resynthesized partial signal p [n-δ_1-δ_2-δ_3] from above. The subtractor 620 outputs a notch-filtered signal resulting from the subtraction of the partial signal from the input signal.

[0029] Referring now to Figure 7, a plurality of frequency locked loop trackers 700-1 to 700-N according to the preferred embodiment of the present invention are servoed in a harmonic locked loop tracker 701. The frequency locked loop tracker of the preferred embodiment of the present invention performs fast and accurate tracking of the instantaneous frequency of a single target partial signal in isolation. However if the signal to noise ratio is large, tracking may break down. Acoustical signals are often composed of complex mixtures of signals which bring the signal to noise ratio for a target partial signal down below the level needed for tracking according to the frequency locked loop method disclosed above. However, the harmonic structure of many natural acoustic signals allows for the robust tracking of the harmonic set of partials associated with a given harmonic signal. Accordingly, a harmonic locked loop tracker 701 is provided wherein a plurality of frequency locked loop trackers are servoed to track a partial signal and a plurality of harmonics where each of the harmonics is a multiple of the fundamental frequency of the partial signal being tracked.

[0030] In the first step of the analysis of a harmonic signal s[n], an instantaneous frequency correction term is calculated for each harmonic. Specifically, the harmonic signal s[n] is demodulated by the frequency warping signal 706 via multipliers 704 for each stage. Each stage further includes a low pass filter 708 which receives the complex demodulated signal d_k[n] which in turn produces an analytic signal u_k[n]. This resultant signal u_k[n] is then combined with a conjugate of itself delayed by one sample via multiplier 710 and delay element 712. The resultant output of the multiplier 710 is coupled to a phase extraction module 714 in order to calculate the phase difference of the resultant signal. The phase extraction module 714 is normalized by combining a normalization signal ($f_s/2\pi k$ where f_s is the sampling frequency) via multiplier 716, resulting in a error term $\varepsilon_{f,o}^{(k)}$[n]. The division by "k" takes into account that the kth stage is tracking "k" times the fundamental frequency.

[0031] In the second step of the analysis, the resulting error signals $\varepsilon_{f,o}^{(k)}$[n] are combined for each stage to yield an overall optimized error correction for use by the frequency estimator and phase accumulator of the frequency locked loop tracker disclosed above. In the preferred embodiment, the frequency corrections from each tracker are weighted in accordance with the inverse of the variance of its tracking performance. Hence each harmonic of the tracked fundamental signal with a low variance will be weighted strongly, while harmonics with high variance (e.g., in noisy portions of the spectrum) will be weighted less strongly. The resultant fundamental frequency estimate is a minimum variance estimate, and is better than the best single frequency locked loop estimate.

[0032] Specifically, the error signal $\varepsilon^{(k)}_{f,o}[n]$ is utilized in order to calculate a variance estimate for each of the individual phase trackers. In each tracker, the error signal $\varepsilon^{(k)}_{f,o}[n]$ is multiplied by itself via squaring module 750. The output of the squaring module 750 is coupled to a variance estimator 752 utilized to calculate the variance of the error signal $\varepsilon^{(k)}_{f,o}[n]$. The variance estimator 752 derives a variance estimate $\overline{\varepsilon^{2}}^{(k)}_{f,o}[n]$ according to the formula

$$\overline{\varepsilon^{2}}^{(k)}_{f,o}[n] = g_k[n]\overline{\varepsilon^{2}}^{(k)}_{f,o}[n-1] + (1-g_k[n])(\varepsilon^{(k)}_{f,o}[n])^2$$

wherein the time constant $g_k[n]$ may be time varying and an exponential weighting scheme is used. Those ordinarily skilled in the art will recognize that other weighting schemes may be utilized in order to determine how the individual phasor signals will be combined in order to optimize partial signal tracking.

[0033] In the preferred embodiment of the present invention, the resultant variance estimate $\overline{\varepsilon^{2}}^{(k)}_{f,o}[n]$ is inverted by module 754 and then coupled to a saturation detector 756. The saturation detector serves to compensate for signals with a high signal to noise ratio for the particular harmonic being tracked. When the signal to noise ratio is too high, the variance estimate becomes limited by the band width of the low pass filter 708 causing it to be too low. When the variance estimate is saturated in this way, it causes the weighting for its associated tracker to be too high. This saturated variance estimate associated with the particular harmonic tracking stage then becomes an unreliable estimator of the true variance of the single target partial p[n] for this particular harmonic. This is especially a problem for higher harmonics where often a mix of broad band noise and audio signals occurs. The weighting given to the particular frequency and phase error associated with the individual harmonic is proportional to the reciprocal of the estimated variance thus not allowing for the higher harmonics to become unfairly highly weighted. In the preferred embodiment, the saturation detector 756 output $w_k[n]$ is defined as

$$w_k[n] = 1/\overline{\varepsilon^{2}}^{(k)}_{f,o}[n] \text{ if } \overline{\varepsilon^{2}}^{(k)}_{f,o}[n] < BW^2/24k^2$$

otherwise $w_k[n] = 1/k^2\overline{\varepsilon^{2}}^{(k)}_{f,o}[n]$

where BW equals the bandwidth of the kth low pass filter 708.

[0034] The output of the saturation detector is combined via multiplier 757 with the individual error signal $\varepsilon^{(k)}_{f,o}[n]$ to yield a weighted phase error signal. Each of the weighted error signals are combined by adders 758 and combined with the sum of the weights from each of the saturation detectors 756 for each harmonic phase tracker. The sum of the weights is inverted prior to combination with the sum of the phase error signals by inverter 760 in order to provide a normalizing factor for the summed phase error signal. The output of the multiplier 762 is the weighted phase error signal which is then combined with the tracker attenuation gain $g_0[n]$ and integrated to produce the estimated fundamental frequency $\hat{f}_o[n]$ for use in the demodulation of the input signal 702 as was described in accordance with the frequency locked loop tracker above.

[0035] Those ordinarily skilled in the art will recognize that any of the number of weighting schemes may be utilized in order to combine the individual phase error signals which result from each harmonic loop tracker. The particular inverse variance method selected should not be construed as limiting.

[0036] The input signal s[n] may include several voices, each comprising a fundamental partial signal and a set corresponding harmonics. The harmonics tracked by the set of parallel trackers in Figure 7 can be resynthesized so as to regenerate one complete "voice". In one preferred embodiment, such resynthesis is accomplished using one instance of the resynthesis module (i.e., multiplier 402) shown in Figure 4 for each of the trackers. Improved resynthesis is accomplished in a second preferred embodiment by providing one instance of the resynthesis module shown in Figure 5 or Figure 6 for each of the trackers in Figure 7.

[0037] Those ordinarily skilled in the art will recognize that the harmonic loop tracker described in the preferred embodiment may also be used for tracking a well defined partial signal along with non-integer multiples of the fundamental frequency. This type of tracking known as inharmonic tracking is especially useful in tracking audio signals such as a piano, wherein sounds emanating from a piano are composed of stretched partials which are not integer multiples of a particular fundamental frequency. Inharmonic tracking is accomplished by defining a constant inharmonic ratio between the kth partial and the fundamental frequency. Such inharmonic frequency ratios may be supplied by a template or may be adaptively trained. In the preferred embodiment, the tracking of the inharmonic partials is the same with the exception that the kth demodulated signal must be computed explicitly, instead of in an iterative cascade, since the partials are no longer integer multiples of the fundamental frequency.

ALTERNATE EMBODIMENTS

[0038] Although the present invention has been described with reference to a few specific embodiments, the foregoing descriptions are illustrative of the invention and should not to be construed as limiting. Various modifications may occur to those skilled in the art without departing from the scope of the invention as defined by the appended claims.

[0039] For instance, the minimum-variance weighting method of the present invention could be used with a set of harmonically constrained peak detectors in an FFT-based pitch tracker.

Claims

1. A frequency-locked loop pitch tracker for tracking an input signal comprising:

demodulation means (104) including a demodulation signal for demodulating said input signal resulting in a complex demodulated signal;
a low pass filter (108) receiving said complex demodulated signal, said low pass filter for producing a filtered analytic signal;
means for detecting (110, 112, 114, 116) the rate of phase change of said filtered analytical signal and for producing a frequency tracking error signal;
an accumulator (120) for receiving said frequency tracking error signal and outputting an estimated input signal frequency; and
means (124) for updating said demodulation signal responsive to said estimated input signal frequency;
said accumulator including an integrator (120) for receiving said frequency tracking error signal and producing said estimated input signal frequency and a frequency-smoothing filter coupled (302, 304, 306) to said integrator for receiving said integrator output signal and thereby improving said outputted estimated input signal frequency.

2. The pitch tracker of claim 1 wherein said demodulation means comprises a multiplier for multiplying said input signal by the complex conjugate of a frequency-warping signal.

3. The pitch tracker of claim 1 or 2, further including means for subtracting a resynthesized partial signal from said input signal, said subtraction means including:

a resynthesizer for resynthesizing a partial signal from said filtered analytic signal and said demodulation signal; and
a subtractor for subtracting said resynthesized partial signal from said input signal.

4. The pitch tracker of claim 1 or 2 further including a resynthesizer, said resynthesizer including multiplier means for combining said demodulation signal with said filtered analytic signal to yield a resynthesized single partial target signal.

5. The pitch tracker of claim 4 further including a subtractor for removing said resynthesized single partial target signal from said input signal, said subtractor including
a delay line for compensating for group delay in said low pass filter resulting in a delayed input signal; and
a subtraction means having first and second inputs and a subtraction output, said subtraction means first input for receiving said delayed input signal and said subtraction means second input for receiving said resynthesized single partial target signal, such that said subtraction means generates a residual signal at said subtraction means output by removing said resynthesized single partial target signal from said delayed input signal.

6. The pitch tracker of claim 5, said resynthesizer including:

a second demodulation means including a second demodulation signal responsive to said improved frequency estimate signal for generating a second complex demodulated signal;
a second delay line for matching the group delays of said low pass filter and a Kay filter, said second delay line coupling said input signal to said second demodulation means;
a second low pass filter receiving said second complex demodulated signal, said second low pass filter for producing a second filtered analytic signal;
a third delay line receiving said second demodulation signal for producing a delayed second demodulation

signal having a delay equal to the group delay of said second low pass filter,
multiplier means for combining said delayed second demodulation signal with said second filtered analytic signal for producing a resynthesized single partial target signal.

7. The pitch tracker of claim 1 or 2, further including phase-locked tracking means, said phase locked tracking means processing said filtered analytic signal using a complex phase detection function and producing a phase error signal, said phase error signal coupled to said means for updating said demodulation signal such that phase-locking is achieved.

8. A frequency-locked loop pitch-tracking method for tracking an input signal comprising the steps of:

demodulating said input signal with a demodulation signal resulting in a complex demodulated signal;
filtering said complex demodulated signal with a low pass filter, said low pass filter for producing a filtered analytic signal;
detecting the rate of phase change of said filtered analytical signal to produce a frequency tracking error signal;
outputting an estimated input signal frequency responsive to said frequency tracking error signal; and
updating said demodulation signal responsive to said estimated input signal frequency;
said outputting step including integrating said frequency tracking error signal to produce said estimated input signal frequency and filtering said integrator output signal with a frequency-smoothing filter to thereby improve said estimated input signal frequency.

9. The method of claim 8, wherein said demodulating step includes multiplying said input signal by a frequency-warping signal's complex conjugate.

10. The method of claim 8 further including combining said complex demodulated signal with said filtered analytic signal to yield a resynthesized single partial target signal.

11. The method of claim 10 further including:

subtracting said resynthesized partial signal from said input signal to generate a residual signal.

12. The method of claim 11, said subtracting step including:

generating a delayed input signal, and
removing said resynthesized single partial target signal from said delayed input signal to as to generate the residual signal.

13. The method of claim 8, further including the steps of:

combining said demodulation signal with said filtered analytic signal to yield a resynthesized single partial target signal;
generating a delayed input signal by delaying said input signal so as to compensate for signal delay associated with said filtering step; and
subtracting said resynthesized single partial target signal from said delayed input signal to generate a residual signal.

14. A pitch tracker for tracking an input signal by tracking a plurality of harmonics in a harmonic signal representation of said input signal comprising:

a like plurality of frequency trackers, each in accordance with any of claims 1 to 7, each of said frequency trackers responsive to an estimated frequency signal for tracking one of said harmonics and producing a frequency tracking error signal; wherein said plurality of frequency trackers are harmonically constrained such that each frequency tracker tracks a respective integer multiple of a fundamental frequency component of said input signal;
means for weighting each of said frequency tracking error signals from each of said plurality of frequency trackers for producing a weighted frequency tracking error signal; and
an accumulator for receiving said weighted frequency tracking error signals and outputting an updated estimated frequency signal such that each said frequency tracker tracks a corresponding one of said harmonics

in accordance with said updated frequency estimate signal.

15. The pitch tracker of claim 14,
each of said frequency trackers including:

demodulation means including a demodulation signal for demodulating said one of said harmonics resulting in a complex demodulated signal;
a low pass filter receiving said complex demodulated signal, said low pass filter for producing a filtered analytic signal; and
means for detecting the rate of phase change of said filtered analytical signal and for producing a frequency tracking error signal;

said pitch tracker further including means for updating said demodulation signal responsive to said estimated input signal frequency.

16. The pitch tracker of claim 13 or 14, wherein
each of said frequency trackers further includes a variance estimator for calculating the variance of said frequency tracking error signal; and
each respective one of said frequency tracking error signals is weighted in accordance with the inverse of the variance of said respective frequency tracking error signal.

17. The pitch tracker of claim 16, wherein said variance estimator derives the variance of said frequency tracking error signal according to the formula:

$$\overline{\varepsilon^2}_k[n] = g_k[n]\overline{\varepsilon^2}_k[n\text{-}1] + (1\text{-}g_k[n])\varepsilon_k^2[n]$$

where
$\overline{\varepsilon^2}_k[n]$ is the variance estimate;
$\varepsilon_k[n]$ is the frequency tracking error signal for *k*th harmonic, and
$g_k[n]$ is the loop gain.

18. The pitch tracker of claim 16, wherein said weighting means further includes a saturation detector to limit the weighting of any frequency estimate due to a kth-tracker in cases where said variance estimate saturates.

19. The method of any of claims 8 to 13, further **characterized by** tracking the input signal by tracking a plurality of harmonics in a harmonic signal representation of said input signal comprising:

a) performing the method of any of claims 8 to 13 using a plurality of frequency trackers, each of said frequency trackers demodulating said input signal with a demodulation signal for tracking one of said harmonics; wherein said plurality of frequency trackers are harmonically constrained such that each frequency tracker tracks a respective integer multiple of a fundamental frequency component of said input signal;
b) deriving a frequency error tracking signal for each of said harmonics;
c) weighting each of said frequency tracking error signals from each of said plurality of frequency trackers for producing a weighted frequency tracking error signal;
d) outputting an estimated input signal frequency responsive to said weighted frequency tracking error signal; and
e) updating said demodulation signal responsive to said estimated input signal frequency.

20. The method of claim 19,
further including the steps of determining the variance of said frequency tracking error signal for each of said harmonics, and determining when said variance estimate saturates;
said weighting step including limiting the weighting of each frequency tracking error signal whose variance estimate saturates.

21. The method of claim 20, wherein the step of determining the variance of said frequency tracking error signal for each of said harmonics, is performed according to the formula:

$$\overline{\varepsilon^2}_k[n] = g_k[n]\overline{\varepsilon^2}_k[n-1] + (1-g_k[n])\varepsilon^2{}_k[n]$$

where

$\overline{\varepsilon^2}_k[n]$ is the variance estimate;
$\varepsilon_k[n]$ is the frequency tracking error signal for kth harmonic, and
$g_k[n]$ is the loop gain.

22. The method of claim 19, said weighting step including:

a) weighting each of said frequency tracking error signals by the reciprocal of said variance determined for each of said frequency tracking error signals; and
b) summing all of the weighted frequency tracking error signals to yield said weighted frequency tracking error signal.

Patentansprüche

1. Frequenzstarre Grundfrequenz-Folgeschaltung zum Verfolgen eines Eingangssignals mit:

einer Demodulationseinrichtung (104) mit einem Demodulationssignal zum Demodulieren des Eingangssignals zu einem komplexen demodulierten Signal, einem Tiefpassfilter (108), der das komplexe demodulierte Signal empfängt und ein gefiltertes analytisches Signal erzeugt,
einer Einrichtung (110, 112, 114, 116) zum Erfassen der Rate der Phasenänderung des gefilterten analytischen Signals und zum Erzeugen eines Frequenzfolge-Fehlersignals,
einem Akkumulator (120) zum Empfangen des Frequenzfolge-Fehlersignals und
Ausgeben einer geschätzten Eingangssignalfrequenz und
einer Einrichtung (124) zum Aktualisieren des Demodulationssignals entsprechend der geschätzten Eingangssignalfrequenz,

wobei der Akkumulator einen Integrator (120) enthält, der das Frequenzfolge-Fehlersignal empfängt und die geschätzte Eingangssignalfrequenz erzeugt, und einen an den Integrator angeschlossenen (302, 304, 306) Frequenzglättungsfilter, der das Ausgangssignal des Integrators empfängt und dadurch die ausgegebene geschätzte Eingangssignalfrequenz verbessert.

2. Grundfrequenz-Folgeschaltung nach Anspruch 1, wobei die Demodulationseinrichtung einen Multiplizierer zum Multiplizieren des Eingangssignals mit dem komplexen Konjugat eines Frequenzverschiebungssignals.

3. Grundfrequenz-Folgeschaltung nach Anspruch 1 oder 2, ferner mit einer Einrichtung zum Subtrahieren eines resynthetisierten partiellen Signals aus dem Eingangssignal,
wobei die Subtrahiereinrichtung enthält:

einen Resynthetisierer zum Resynthetisieren eines partiellen Signals aus dem gefilterten analytischen Signal und dem Demodulationssignal und
eine Subtrahiereinrichtung zum Subtrahieren des resynthetisierten partiellen Signals vom Eingangssignal.

4. Grundfrequenz-Folgeschaltung nach Anspruch 1 oder 2, ferner mit einem Resynthetisierer mit einer Multipliziereinrichtung zum Kombinieren des Demodulationssignals mit dem gefilterten analytischen Signal zu einem resynthetisierten einzelnen partiellen Zielsignal.

5. Grundfrequenz-Folgeschaltung nach Anspruch 4, ferner mit einer Subtrahiervorrichtung zum Entfernen des resynthetisierten einzelnen partiellen Zielsignals vom Eingangssignal, wobei die Subtrahiervorrichtung enthält:

eine Verzögerungsleitung zum Kompensieren der Gruppenverzögerung im Tiefpassfilter, so dass ein verzögertes Eingangssignal entsteht und
eine Subtrahiereinrichtung mit einem ersten und einem zweiten Eingang und einem Subtraktionsausgang, wobei dem ersten Eingang das verzögerte Eingangssignal und
dem zweiten Eingang das resynthetisierte einzelne partielle Zielsignal zugeführt wird, so dass die Subtrakti-

onseinrichtung ein Restsignal an ihrem Ausgang erzeugt, indem das resynthetisierte einzelne partielle Zielsignal vom verzögerten Eingangssignal entfernt wird.

6. Grundfrequenz-Folgeschaltung nach Anspruch 5, wobei der Resynthetisierer enthält:

eine zweite Demodulationseinrichtung mit einem zweiten Demodulationssignal, die auf das verbesserte Frequenzschätzungssignal anspricht und ein zweites komplexes Demoduliersignal erzeugt,
eine zweite Verzögerungsleitung zum Anpassen der Gruppenverzögerungen des Tiefpassfilters und eines KAY-Filters, wobei die zweite Verzögerungsleitung das Eingangssignal auf die zweite Demodulationseinrichtung führt,
einen zweiten Tiefpassfilter, der das zweite komplexe demodulierte Signal empfängt und ein zweites gefiltertes analytisches Signal erzeugt,
eine dritte Verzögerungsleitung, die das zweite Demodulationssignal empfängt und
ein verzögertes zweites Demodulationssignal erzeugt, dessen Verzögerung gleich der Gruppenverzögerung des zweiten Tiefpassfilters ist,
eine Multipliziereinrichtung zum Kombinieren des verzögerten zweiten Demodulationssignals mit dem zweiten gefilterten analytischen Signal zum Erzeugen eines resynthetisierten einzelnen partiellen Zielsignals.

7. Grundfrequenz-Folgeschaltung nach Anspruch 1 oder 2, ferner mit einer phasenstarren Folgeeinrichtung, die das gefilterte analytische Signal unter Verwendung einer komplexen Phasenerfassungsfunktion verarbeitet und ein Phasenfehlersignal erzeugt, das der Einrichtung zum Aktualisieren des Demodulationssignals zugeführt wird, so dass eine Phasenverriegelung erzielt wird.

8. Frequenzstarres Grundfrequenz-Folgeverfahren zum Verfolgen eines Eingangssignals mit folgenden Schritten:

Demodulieren des Eingangssignals mit einem Demodulationssignal zu einem komplexen demodulierten Signal,
Filtern des komplexen demodulierten Signals mit einem Tiefpassfilter, der ein gefiltertes analytisches Signal erzeugt,
Erfassen der Rate der Phasenänderung des gefilterten analytischen Signals zur Erzeugung eines Frequenzfolge-Fehlersignals,
Ausgeben einer geschätzten Eingangssignalfrequenz auf das Frequenzfolge-Fehlersignal und
Aktualisieren des Demodulationssignals auf die geschätzte Eingangssignalfrequenz,

wobei beim Ausgeben das Frequenzfolge-Fehlersignal integriert und die geschätzte Eingangssignalfrequenz erzeugt und das Ausgangssignal des Integrators mit einem Frequenz-Glättungsfilter gefiltert wird, um die geschätzte Eingangssignalfrequenz zu verbessern.

9. Verfahren nach Anspruch 8, wobei der Demodulationsschritt das Multiplizieren des Eingangssignals mit einem komplexen Konjugat eines Frequenzverschiebungssignals umfasst.

10. Verfahren nach Anspruch 8, wobei das komplexe demodulierte Signal mit dem gefilterten analytischen Signal kombiniert wird, so dass sich ein resynthetisiertes einzelnes partielles Zielsignal ergibt.

11. Verfahren nach Anspruch 10, wobei das resynthetisierte partielle Signal vom Eingangssignal subtrahiert und ein Restsignal erzeugt wird.

12. Verfahren nach Anspruch 11, wobei der Subtraktionsschritt umfasst:

Erzeugen eines verzögerten Eingangssignals und
Entfernen des resynthetisierten einzelnen partiellen Zielsignals vom verzögerten Eingangssignal, so dass das Restsignal erzeugt wird.

13. Verfahren nach Anspruch 8, ferner mit folgenden Schritten:

Kombinieren des Demodulationssignals mit dem gefilterten analytischen Signal zu einem resynthetisierten einzelnen partiellen Zielsignal,
Erzeugen eines verzögerten Eingangssignals durch Verzögern des Eingangssignals so, dass die mit dem

Filtrierschritt verbundene Signalverzögerung kompensiert wird und
Abziehen des resynthetisierten einzelnen partiellen Signals vom verzögerten Eingangssignal zur Erzeugung eines Restsignals.

14. Grundfrequenz-Folgeschaltung zum Verfolgen eines Eingangssignals durch Verfolgen einer Anzahl von Harmonischen in einer harmonischen Signaldarstellung des Eingangssignals mit
Einer gleichen Anzahl von Frequenz-Folgeschaltungen nach einem der Ansprüche 1 bis 7, wobei jede Frequenzfolgeschaltung auf ein geschätztes Frequenzsignal anspricht, eine der Harmonischen verfolgt und ein Frequenzfolge-Fehlersignal erzeugt, wobei die Frequenzfolgeschaltungen harmonisch beschränkt sind, so dass jede Frequenzfolgeschaltung ein entsprechendes ganzzahliges Vielfach einer Grundfrequenzkomponente des Eingangssignals verfolgt,
Einrichtungen zum Gewichten jedes Frequenzfolge-Fehlersignals von jedem der Frequenzfolgeschaltungen zum Erzeugen eines gewichteten Frequenzfolge-Fehlersignals und
einem Akkumulator zum Empfangen der gewichteten Frequenzfolge-Fehlersignale und Ausgeben eines aktualisierten geschätzten Frequenzsignals, so dass jeder der Frequenzfolgeschaltungen eine entsprechende der Harmonischen entsprechend dem aktualisierten Frequenzschätzungsignal verfolgt.

15. Grundfrequenz-Folgungsschaltung nach Anspruch 14, wobei jede der Frequenzfolgeschaltungen enthält

eine Demodulationseinrichtung mit einem Demodulationssignal zum Demodulieren der einen Harmonischen zu einem komplexen demodulierten Signal,
ein Tiefpassfilter, das das komplexe demodulierte Signal empfängt und ein gefiltertes analytisches Signal erzeugt und
eine Einrichtung zum Erfassen der Rate der Phasenänderung des gefilterten analytischen Signals und Erzeugen eines Frequenzfolge-Fehlersignals

wobei die Grundfrequenz-Folgeschaltung ferner eine Einrichtung zum Aktualisieren des Demodulationssignals auf die geschätzte Eingangssignalfrequenz enthält.

16. Grundfrequenz-Folgeschaltung nach Anspruch 13 oder 14, wobei jede der Frequenzfolgeschaltungen ferner einen Veränderlichkeitsschätzer zum Berechnen der Veränderlichkeit des Frequenzfolge-Fehlersignals enthält und jedes der Frequenzfolge-Fehlersignale entsprechend der inversen Veränderlichkeit des jeweiligen Frequenzfolgefehlersignals gewichtet wird.

17. Grundfrequenz-Folgeschaltung nach Anspruch 16, wobei der Veränderlichkeitsschätzer die Veränderlichkeit des Frequenzfolge-Fehlersignals entsprechend folgender Gleichung berechnet:

$$\overline{\varepsilon^2}_k[n] = g_k[n]\overline{\varepsilon^2}_k[n\text{-}1] + (1\text{-}g_k[n])\varepsilon^2{}_k[n]$$

worin
$\bar{\varepsilon}^2{}_k[n]$ die Veränderlichkeitsschätzung,
$\varepsilon_k[n]$ das Frequenzfolge-Fehlersignal für die k-te Harmonische und
$g_k[n]$ die Schaltungsverstärkung.

18. Grundfrequenz-Folgeschaltung nach Anspruch 16, wobei die Gewichtungseinrichtung ferner einen Sättigungsdetektor enthält, zum Begrenzen der Gewichtung jeglicher Frequenzschätzung infolge einer k-ten Folgeschaltung in Fällen, in denen sich die Veränderlichkeitsschätzung sättigt.

19. Verfahren nach einem der Ansprüche 8 bis 13, ferner **gekennzeichnet durch** Verfolgen des Eingangssignals **durch** Verfolgen einer Anzahl von Harmonischen in einer harmonischen Signaldarstellung des Eingangssignals mit folgenden Schritten:

a) Durchführen des Verfahren nach einem der Ansprüche 8 bis 13 unter Verwendung einer Anzahl von Frequenzfolgeschaltungen, die je das Eingangssignal mit einem Demodulationssignal demodulieren, zum Verfolgen einer der Harmonischen, wobei die Anzahl der Frequenzfolgeschaltungen harmonisch beschränkt ist, so dass jede Frequenzfolgeschaltung jeweils ein ganzzahliges Vielfaches einer Grundfrequenzkomponente des Eingangssignals verfolgt,

b) Berechnen eines Frequenzfolge-Fehlersignals für jede der Harmonischen,
c) Gewichten jedes Frequenzfolge-Fehlersignals von jedem der Anzahl von Frequenzfolgeschaltungen zum Erzeugen eines gewichteten Frequenzfolge-Fehlersignals,
d) Ausgeben einer geschätzten Signalfrequenz auf das gewichtete Frequenzfolge-Fehlersignal und
e) Aktualisieren des Demodulationssignals auf die geschätzte Eingangssignalfrequenz.

20. Verfahren nach Anspruch 19 ferner mit den Schritten des Bestimmens der Veränderlichkeit des Frequenzfehler-Folgesignals für jede der Harmonischen und Bestimmen, wenn sich die Veränderlichkeitsschätzung sättigt, wobei der Gewichtungsschritt die Begrenzung der Gewichtung jedes Frequenzfolge-Fehlersignals umfasst, dessen Veränderlichkeitsschätzung in die Sättigung geht.

21. Verfahren nach Anspruch 20, wobei der Schritt des Bestimmens der Veränderlichkeit des Frequenzfolge-Fehlersignals für jede Harmonische nach folgender Gleichung erfolgt:

$$\overline{\varepsilon^2}_k[n] = g_k[n]\overline{\varepsilon^2}_k[n\text{-}1] + (1\text{-}g_k[n])\varepsilon^2{}_k[n]$$

worin
$\overline{\varepsilon^2}_k[n]$ die Veränderlichkeitsschätzung,
$\varepsilon_k[n]$ das Frequenzfolge-Fehlersignal für die k-te Harmonische und
$g_k[n]$ die Schaltverstärkung.

22. Verfahren nach Anspruch 19, wobei der Gewichtungsschritt umfasst:

a) Gewichten jedes Frequenzfolge-Fehlersignals durch den reziproken Wert der für jedes der Frequenzfolge-Fehlersignale bestimmten Veränderlichkeit und
b) Summieren aller gewichteten Frequenzfolge-Fehlersignale zu dem gewichteten Frequenzfolge-Fehlersignal.

Revendications

1. Dispositif de recherche de hauteur d'un son à boucle d'asservissement en fréquence pour la recherche d'un signal d'entrée comportant :

un moyen de démodulation (104) comprenant un signal de démodulation pour démoduler ledit signal d'entrée, donnant un signal démodulé complexe ;
un filtre passe-bas (108) recevant ledit signal démodulé complexe, ledit filtre passe-bas étant destiné à produire un signal analytique filtré ;
un moyen (110, 112, 114, 116) destiné à détecter le rythme de variation de phase dudit signal analytique filtré et à produire un signal d'erreur de recherche de fréquence ;
un accumulateur (120) destiné à recevoir ledit signal d'erreur de recherche de fréquence et à délivrer en sortie une fréquence estimée du signal d'entrée ; et
un moyen (124) destiné à mettre à jour ledit signal de démodulation en réponse à ladite fréquence estimée du signal d'entrée ;
ledit accumulateur comprenant un intégrateur (120) destiné à recevoir ledit signal d'erreur de recherche de fréquence et à produire ladite fréquence estimée du signal d'entrée et un filtre (302, 304, 306) de lissage de fréquence couplé audit intégrateur pour recevoir ledit signal de sortie de l'intégrateur et améliorer ainsi ladite fréquence estimée du signal d'entrée délivrée en sortie.

2. Dispositif de recherche de hauteur de son selon la revendication 1, dans lequel ledit moyen de démodulation comporte un multiplicateur destiné à multiplier ledit signal d'entrée par le conjugué complexe d'un signal de distorsion de fréquence.

3. Dispositif de recherche de hauteur de son selon la revendication 1 ou 2, comprenant en outre un moyen destiné à soustraire un signal partiel resynthétisé dudit signal d'entrée, ledit moyen de soustraction comprenant :

un resynthétiseur destiné à resynthétiser un signal partiel à partir dudit signal analytique filtré et dudit signal

de démodulation ; et
un soustracteur destiné à soustraire ledit signal partiel resynthétisé dudit signal d'entrée.

4. Dispositif de recherche de hauteur de son selon la revendication 1 ou 2, comprenant en outre un resynthétiseur, ledit resynthétiseur comprenant un moyen multiplicateur destiné à combiner ledit signal de démodulation avec ledit signal analytique filtré pour donner un signal cible partiel unique resynthétisé.

5. Dispositif de recherche de hauteur de son selon la revendication 4, comprenant en outre un soustracteur destiné à enlever ledit signal cible partiel unique resynthétisé dudit signal d'entrée, ledit soustracteur comprenant
une ligne à retard destinée à compenser un retard de groupe dans ledit filtre passe-bas et, donnant un signal d'entrée retardé ; et
un moyen de soustraction ayant des première et seconde entrées et une sortie de soustraction, ladite première entrée du moyen de soustraction étant destinée à recevoir ledit signal d'entrée retardé et ladite seconde entrée du moyen de soustraction étant destinée à recevoir ledit signal cible partiel unique resynthétisé, de façon que ledit moyen de soustraction génère un signal résiduel à ladite sortie du moyen de soustraction en enlevant ledit signal cible partiel unique resynthétisé dudit signal d'entrée retardé.

6. Dispositif de recherche de hauteur de son selon la revendication 5, ledit resynthétiseur comprenant :

un second moyen de démodulation comprenant un second signal de démodulation en réponse audit signal d'estimation de fréquence amélioré pour générer un second signal démodulé complexe ;
une seconde ligne à retard destinée à adapter les retards de groupe dudit filtre passe-bas et d'un filtre de Kay, ladite seconde ligne à retard couplant ledit signal d'entrée audit second moyen de démodulation ;
un second filtre passe-bas recevant ledit signal démodulé complexe, ledit second filtre passe-bas étant destiné à produire un second signal analytique filtré ;
une troisième ligne à retard recevant ledit second signal de démodulation pour produire un second signal de démodulation retardé ayant un retard égal au retard de groupe dudit second filtre passe-bas ;
un moyen multiplicateur destiné à combiner ledit second signal de démodulation retardé audit second signal analytique filtré pour produire un signal cible partiel unique resynthétisé.

7. Dispositif de recherche de hauteur de son selon la revendication 1 ou 2, comprenant en outre un moyen de recherche asservi en phase, ledit moyen de recherche asservi en phase traitant ledit signal analytique filtré en utilisant une fonction de détection de phase complexe et produisant un signal d'erreur de phase, ledit signal d'erreur de phase étant couplé audit moyen pour mettre à jour ledit signal de démodulation de façon qu'un asservissement de phase soit réalisé.

8. Procédé de recherche de hauteur de son par boucle d'asservissement en fréquence pour rechercher un signal d'entrée, comprenant les étapes qui consistent :

à démoduler ledit signal d'entrée avec un signal de démodulation, donnant un signal démodulé complexe ;
à filtrer ledit signal démodulé complexe avec un filtre passe-bas, ledit filtre passe-bas étant destiné à produire un signal analytique filtré ;
à détecter le rythme de variation de phase dudit signal analytique filtré pour produire un signal d'erreur de recherche de fréquence ;
à délivrer en sortie une fréquence estimée du signal d'entrée en réponse audit signal d'erreur de recherche de fréquence ;
à mettre à jour ledit signal de démodulation en réponse à ladite fréquence estimée du signal d'entrée ;
ladite étape de délivrance en sortie comprenant l'intégration dudit signal d'erreur de recherche de fréquence pour produire ladite fréquence estimée du signal d'entrée et le filtrage dudit signal de sortie de l'intégrateur avec un filtre de lissage de fréquence pour améliorer ainsi ladite fréquence estimée du signal d'entrée.

9. Procédé selon la revendication 8, dans lequel ladite étape de démodulation comprend la multiplication dudit signal d'entrée par un conjugué complexe d'un signal de distorsion de fréquence.

10. Procédé selon la revendication 8, comprenant en outre la combinaison dudit signal démodulé complexe avec ledit signal analytique filtré pour donner un signal cible partiel unique resynthétisé.

11. Procédé selon la revendication 10, comprenant en outre :

la soustraction dudit signal partiel resynthétisé dudit signal d'entrée pour générer un signal résiduel.

12. Procédé selon la revendication 11, ladite étape de soustraction comprenant :

la génération d'un signal d'entrée retardé,
l'enlèvement dudit signal cible partiel unique resynthétisé dudit signal d'entrée retardé afin de générer le signal résiduel.

13. Procédé selon la revendication 8, comprenant en outre les étapes qui consistent :

à combiner ledit signal de démodulation avec ledit signal analytique filtré pour donner un signal cible partiel unique resynthétisé ;
à générer un signal d'entrée retardé en retardant ledit signal d'entrée afin de compenser un retard du signal associé à ladite étape de filtrage ; et
à soustraire ledit signal cible partiel unique resynthétisé dudit signal d'entrée retardé pour générer un signal résiduel.

14. Dispositif de recherche de hauteur de son pour rechercher un signal d'entrée en recherchant une pluralité d'harmoniques dans une représentation d'un signal d'harmonique dudit signal d'entrée, comportant :

une pluralité analogue de dispositifs de recherche de fréquence, chacun conforme à l'une quelconque des revendications 1 à 7, chacun desdits dispositifs de recherche de fréquence réagissant à un signal de fréquence estimée en recherchant l'un desdits harmoniques et en produisant un signal d'erreur de recherche de fréquence ;

dans lequel ladite pluralité de dispositifs de recherche de fréquence est limitée en ce qui concerne les harmoniques de façon que chaque dispositif de recherche de fréquence recherche un multiple entier respectif d'une composante de fréquence fondamentale dudit signal d'entrée ;

un moyen destiné à pondérer chacun desdits signaux d'erreur de recherche de fréquence provenant de chacun de ladite pluralité de dispositifs de recherche de fréquence pour produire un signal d'erreur de recherche de fréquence pondéré ;

un accumulateur destiné à recevoir lesdits signaux d'erreur de recherche de fréquence pondérés et à délivrer en sortie un signal de fréquence estimée mis à jour de façon que chacun desdits dispositifs de recherche de fréquence recherche l'un, correspondant, desdits harmoniques conformément audit signal d'estimation de fréquence mis à jour.

15. Dispositif de recherche de hauteur de son selon la revendication 14,
chacun desdits dispositifs de recherche de fréquence comprenant :

un moyen de démodulation comprenant un signal de démodulation pour démoduler ledit, un, desdits harmoniques, donnant un signal démodulé complexe ;
un filtre passe-bas recevant ledit signal démodulé complexe, ledit filtre passe-bas étant destiné à produire un signal analytique filtré ; et
un moyen destiné à détecter le rythme de variation de phase dudit signal analytique filtré et à produire un signal d'erreur de recherche de fréquence ;
ledit dispositif de recherche de hauteur de son comprenant en outre un moyen destiné à mettre à jour ledit signal de démodulation en réponse à ladite fréquence estimée du signal d'entrée.

16. Dispositif de recherche de hauteur de son selon la revendication 13 ou 14, dans lequel

chacun desdits dispositifs de recherche de fréquence comprend en outre un estimateur de variance destiné à calculer la variance dudit signal d'erreur de recherche de fréquence ; et

chacun, respectif, desdits signaux de recherche de fréquence est pondéré conformément à l'inverse de la variance dudit signal respectif d'erreur de recherche de fréquence.

17. Dispositif de recherche de hauteur de son selon la revendication 16, dans lequel ledit estimateur de variance dérive la variance dudit signal d'erreur de recherche de fréquence conformément à la formule :

A

$$\overline{\varepsilon^2}_k[n] = g_k[n]\overline{\varepsilon^2}_k[n-1] + (1-g_k[n])\varepsilon^2{}_k[n]$$

où

B $\overline{\varepsilon^2}_k[n]$ est l'estimation de la variance ;

C $\varepsilon_k[n]$ est le signal d'erreur de recherche de fréquence pour le kième harmonique, et

D $g_k[n]$ est le gain de la boucle.

18. Dispositif de recherche de hauteur de son selon la revendication 16, dans lequel ledit moyen de pondération comprend en outre un détecteur de saturation destiné à limiter la pondération de toute estimation de fréquence due à un kième dispositif de recherche dans des cas où ladite estimation de la variance est à saturation.

19. Procédé selon l'une quelconque des revendications 8 à 13, **caractérisé en outre par** la recherche du signal d'entrée en recherchant une pluralité d'harmoniques dans une représentation de signal d'harmonique dudit signal d'entrée, comprenant :

a) l'exécution du procédé selon l'une quelconque des revendications 8 à 13 en utilisant une pluralité de, dispositifs de recherche de fréquence, chacun desdits dispositifs de recherche de fréquence démodulant ledit signal d'entrée avec un signal de démodulation pour rechercher l'un desdits harmoniques ; dans lequel ladite pluralité de dispositifs de recherche de fréquence est limitée en ce qui concerne les harmoniques de façon que chaque dispositif de recherche de fréquence recherche un multiple entier respectif d'une composante de fréquence fondamentale dudit signal d'entrée ;
b) la dérivation d'un signal de recherche d'erreur de fréquence pour chacun desdits harmoniques ;
c) la pondération de chacun desdits signaux d'erreur de recherche de fréquence provenant de chacun de ladite pluralité de dispositifs de recherche de fréquence pour produire un signal d'erreur de recherche de fréquence pondéré ;
d) la délivrance en sortie d'une fréquence estimée du signal d'entrée en réponse audit signal d'erreur de recherche de fréquence pondéré ; et
e) la mise à jour dudit signal de démodulation en réponse à ladite fréquence estimée du signal d'entrée.

20. Procédé selon la revendication 19,
comprenant en outre les étapes qui consistent à déterminer la variance dudit signal d'erreur de recherche de fréquence pour chacun desdits harmoniques, et à déterminer lorsque ladite estimation de la variance est à saturation ;
ladite étape de pondération comprenant la limitation de la pondération de chaque signal d'erreur de recherche de fréquence dont l'estimation de la variance est à saturation.

21. Procédé selon la revendication 20, dans lequel l'étape de détermination de la variance dudit signal d'erreur de recherche de fréquence pour chacun desdits harmoniques est effectuée conformément à la formule :

A

$$\overline{\varepsilon^2}_k[n] = g_k[n]\overline{\varepsilon^2}_k[n-1]+(1-g_k[n])\varepsilon^2{}_k[n]$$

où

B $\overline{\varepsilon^2}_k[n]$ est l'estimation de la variance ;

C $\bar{\varepsilon}_k[n]$ est le signal d'erreur de recherche de fréquence pour le kième harmonique, et

D $g_k[n]$ est le gain de la boucle.

22. Procédé selon la revendication 19, dans lequel ladite étape de pondération comprend :

a) la pondération de chacun desdits signaux d'erreur de recherche de fréquence par l'inverse de ladite variance déterminée pour chacun desdits signaux d'erreur de recherche de fréquence ; et
b) la sommation de tous les signaux d'erreur de recherche de fréquence pondérés pour donner ledit signal d'erreur de recherche de fréquence pondéré.

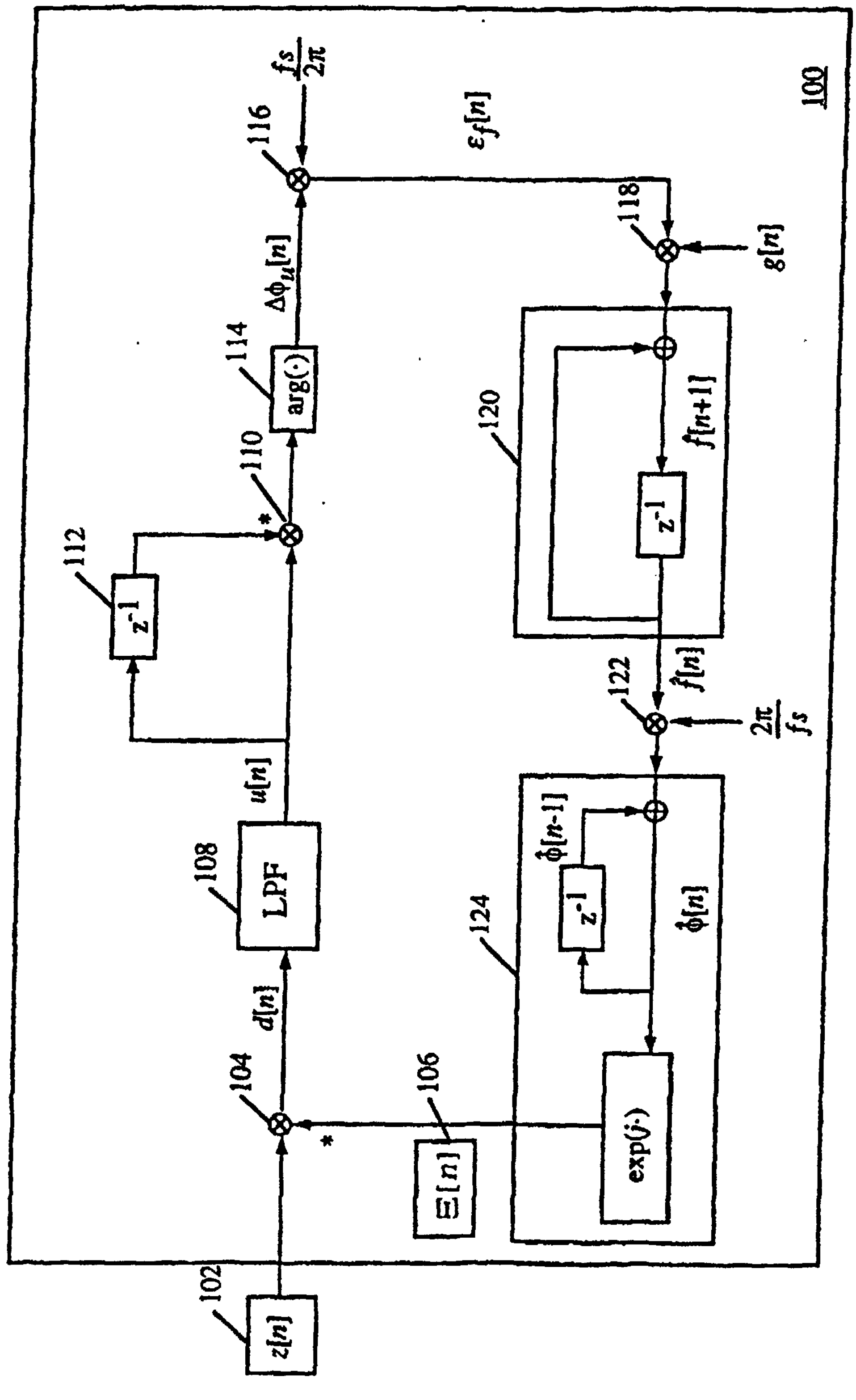

100
102
$z[n]$
104
$d[n]$
106
$\Xi[n]$
108
LPF
$u[n]$
112
z^{-1}
110
114
arg(·)
$\Delta\phi_u[n]$
116
$\frac{fs}{2\pi}$
$\varepsilon_f[n]$
118
$g[n]$
120
$f[n+1]$
z^{-1}
122
$f[n]$
$\frac{2\pi}{fs}$
124
$\hat{\phi}[n-1]$
z^{-1}
$\hat{\phi}[n]$
exp(j·)

FIGURE 1

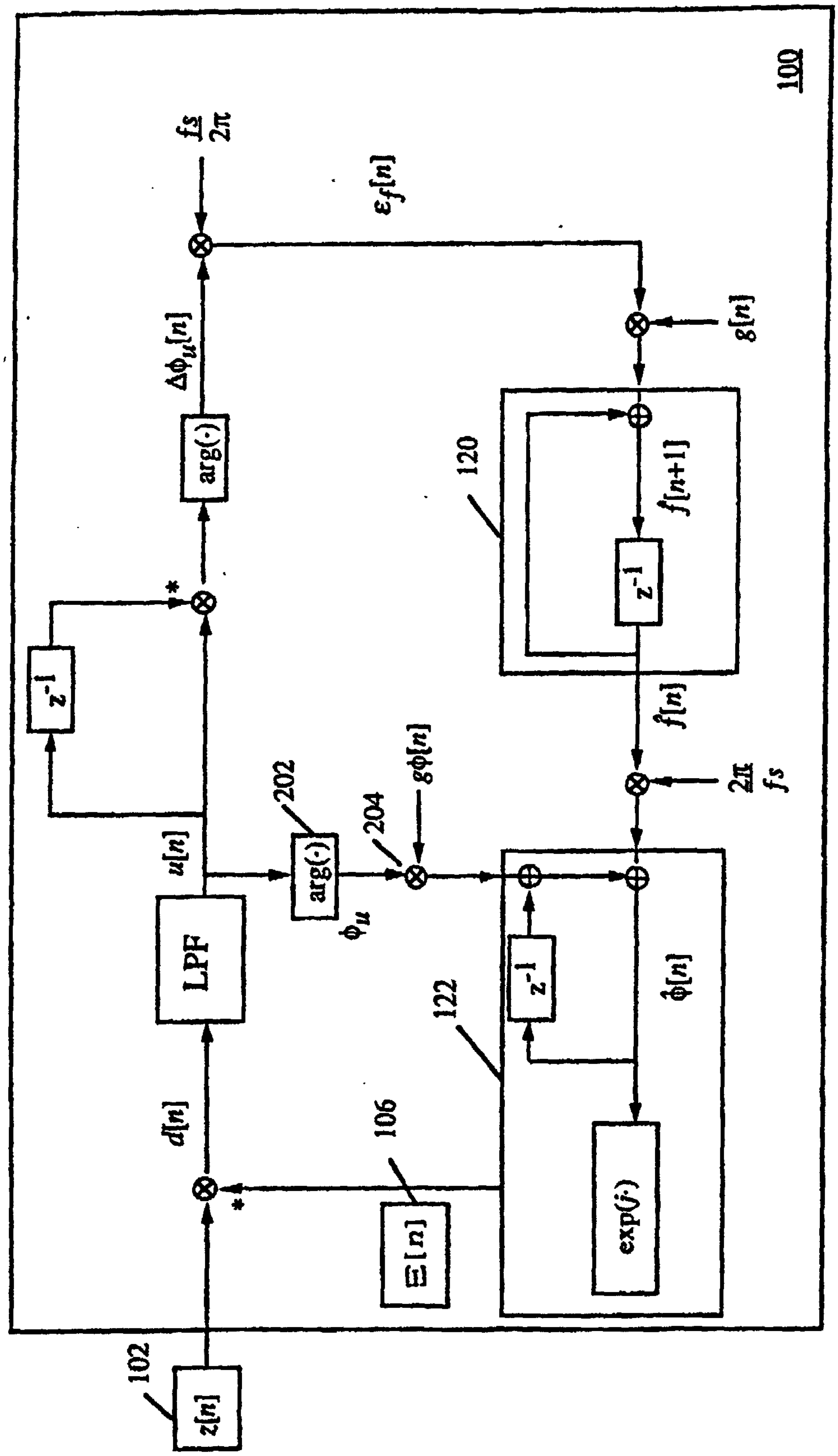

100
z[n]
102
d[n]
LPF
u[n]
z^-1
arg(·)
Δφu[n]
fs/2π
εf[n]
g[n]
120
f[n+1]
f[n]
2π/fs
202
204
gφ[n]
φu
122
φ[n]
106
Ξ[n]
exp(j·)

FIGURE 2

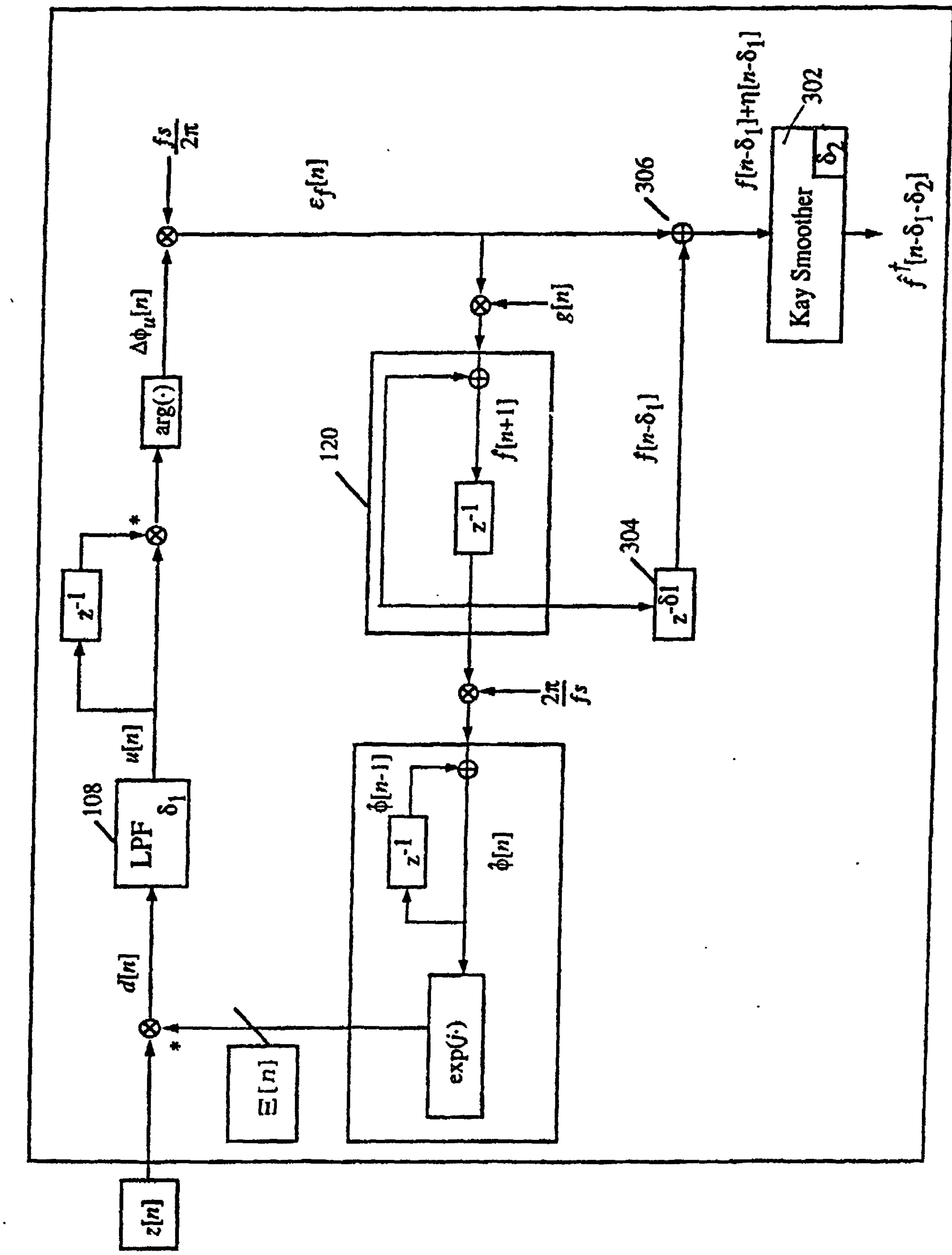

z[n]
d[n]
*
108
LPF
δ1
u[n]
z-1
*
arg(·)
Δφu[n]
fs/2π
εf[n]
g[n]
120
f[n+1]
z-1
2π/fs
φ̂[n-1]
z-1
φ̂[n]
exp(j·)
Ξ[n]
304
z-δ1
f[n-δ1]
306
f[n-δ1]+η[n-δ1]
302
δ2
Kay Smoother
f†[n-δ1-δ2]

FIGURE 3

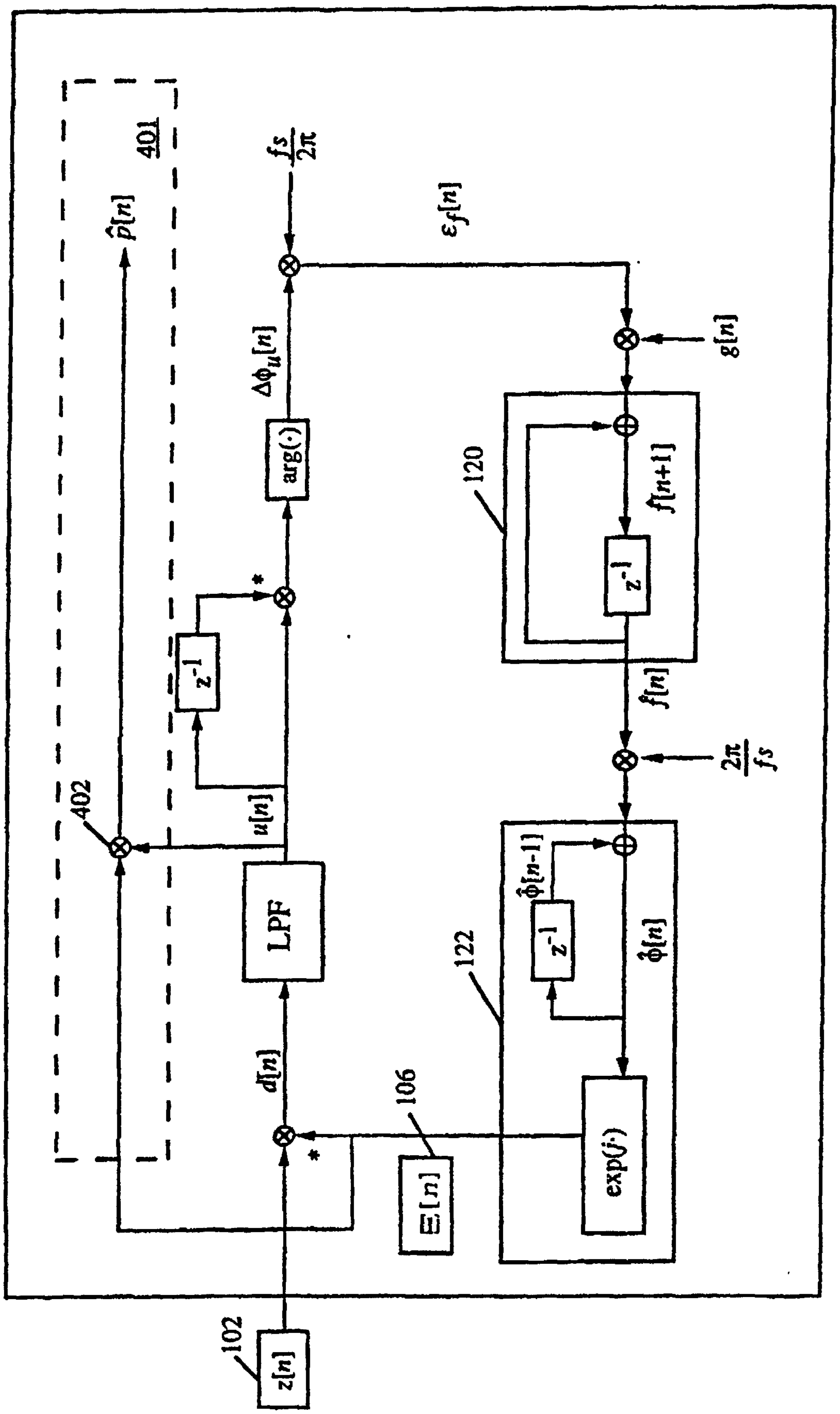
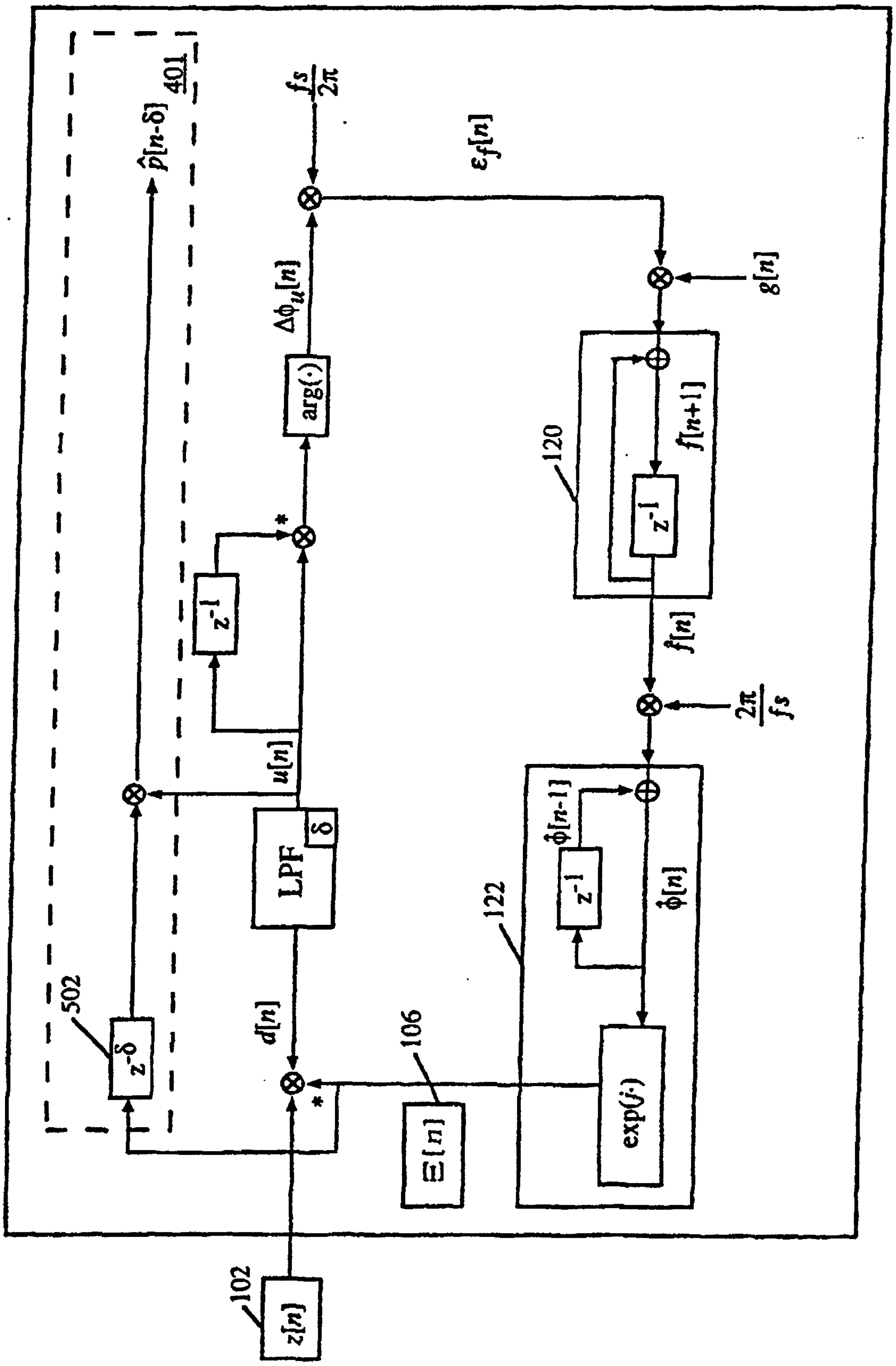

401
402
$\hat{p}[n]$
$u[n]$
z^{-1}
LPF
$d[n]$
106
$\Xi[n]$
102
$z[n]$
arg(·)
$\Delta\Phi_u[n]$
$\frac{fs}{2\pi}$
$\varepsilon_f[n]$
$g[n]$
120
$f[n+1]$
$f[n]$
$\frac{2\pi}{fs}$
122
$\hat{\phi}[n-1]$
$\hat{\phi}[n]$
exp(j·)

FIGURE 4



401
502
$z^{-\delta}$
$\hat{p}[n-\delta]$
$u[n]$
$d[n]$
LPF
δ
z^{-1}
$\arg(\cdot)$
$\Delta\phi_u[n]$
$\frac{fs}{2\pi}$
$\varepsilon_f[n]$
$g[n]$
120
$f[n+1]$
$f[n]$
$\frac{2\pi}{fs}$
122
$\hat{\phi}[n-1]$
$\hat{\phi}[n]$
$\exp(j\cdot)$
106
$\Xi[n]$
102
$z[n]$

FIGURE 5A

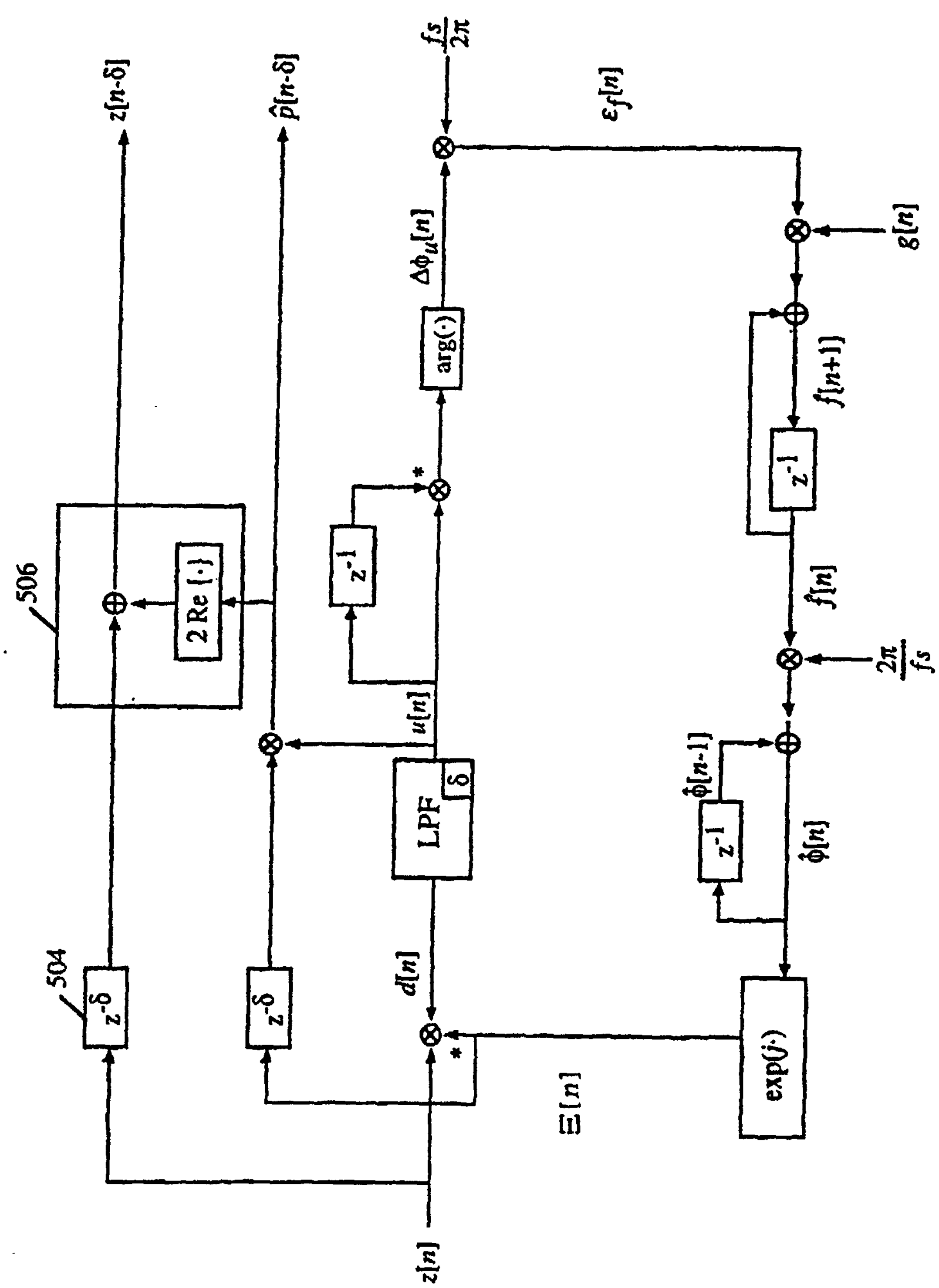

z[n-δ]
p̂[n-δ]
fs/2π
ε_f[n]
Δφ_u[n]
arg(·)
g[n]
f[n+1]
z^-1
f[n]
2π/fs
φ̂[n-1]
φ̂[n]
exp(j·)
Ξ[n]
LPF
δ
u[n]
d[n]
z^-δ
504
506
2 Re{·}
z[n]

FIGURE 5B

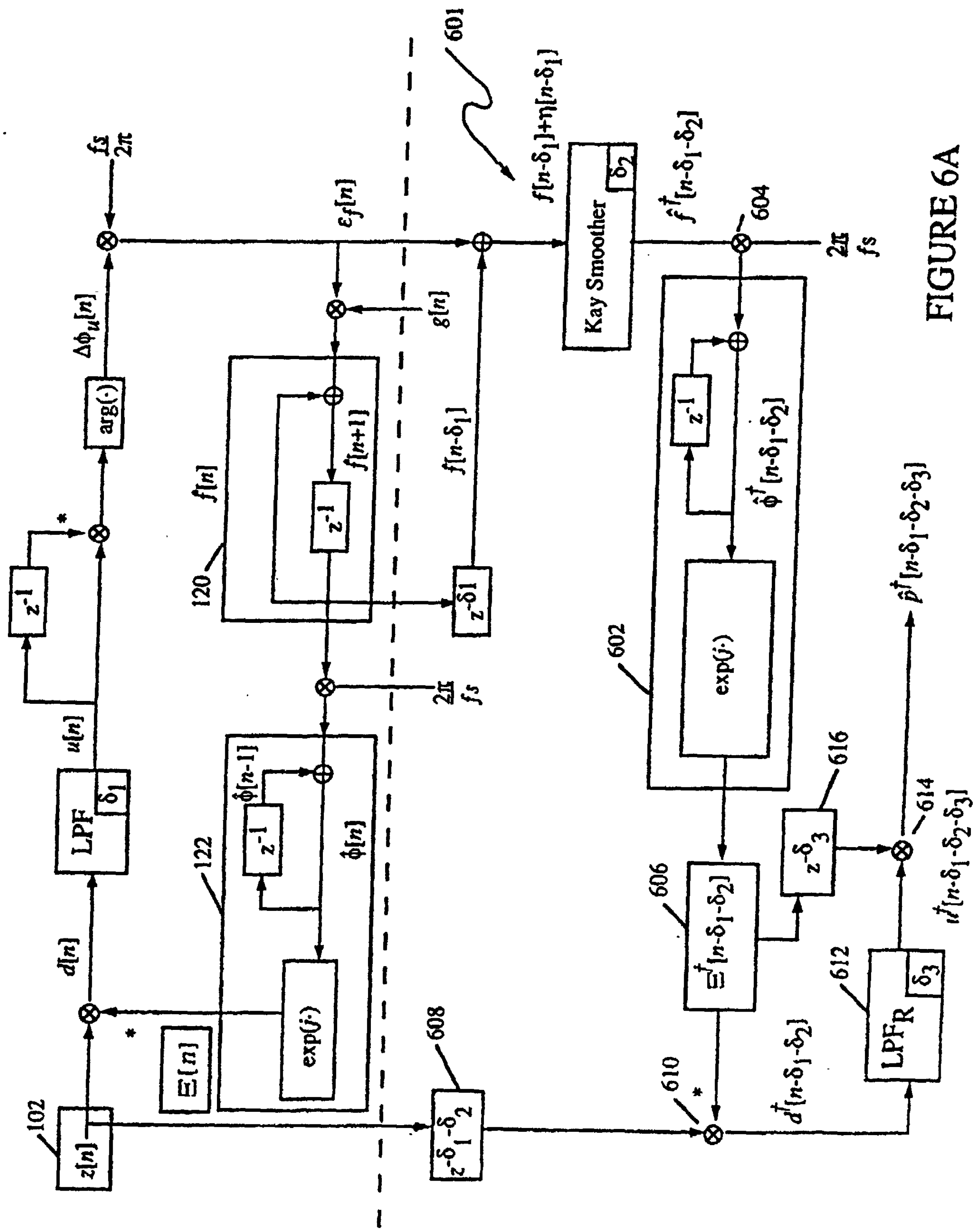

z[n]
LPF
δ_1
arg(·)
z^{-1}
exp(j·)
Kay Smoother
δ_2
LPF_R
δ_3
$\Xi[n]$
601
602
604
606
608
610
612
614
616
102
120
122

FIGURE 6A

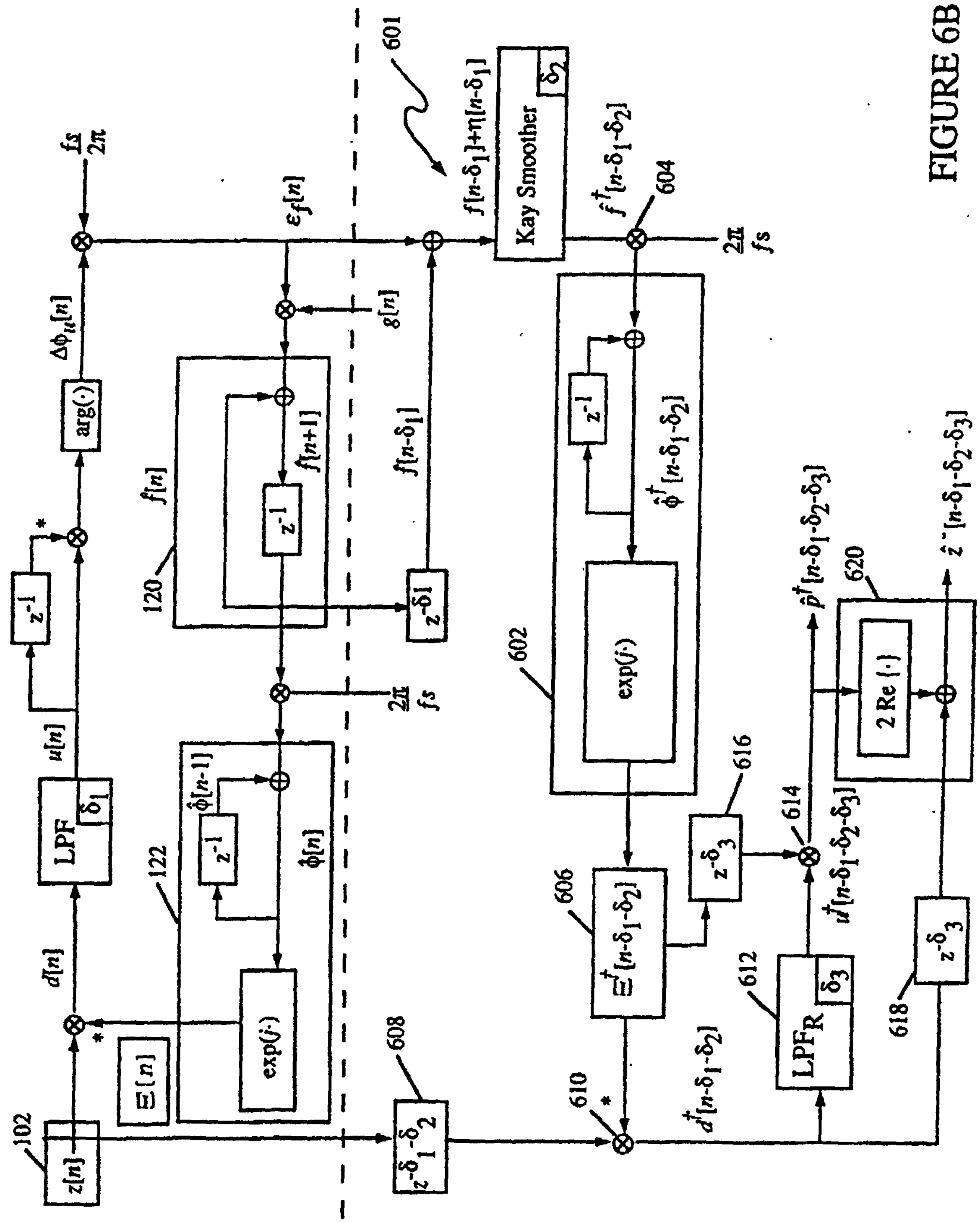

z[n]
102
*
d[n]
LPF
δ1
u[n]
z-1
*
arg(·)
Δφu[n]
fs/2π
εf[n]
g[n]
f[n]
f[n+1]
z-1
120
122
Ξ[n]
exp(j·)
z-1
φ̂[n-1]
φ̂[n]
2π/fs
f[n-δ1]
z-δ1
601
f[n-δ1]+η[n-δ1]
Kay Smoother
δ2
f†[n-δ1-δ2]
604
2π/fs
z-1
φ̂†[n-δ1-δ2]
exp(j·)
602
608
z-δ1-δ2
606
Ξ†[n-δ1-δ2]
z-δ3
616
610
*
d†[n-δ1-δ2]
614
u†[n-δ1-δ2-δ3]
612
LPFR
δ3
618
z-δ3
p̂†[n-δ1-δ2-δ3]
620
2 Re{·}
ẑ-[n-δ1-δ2-δ3]

FIGURE 6B

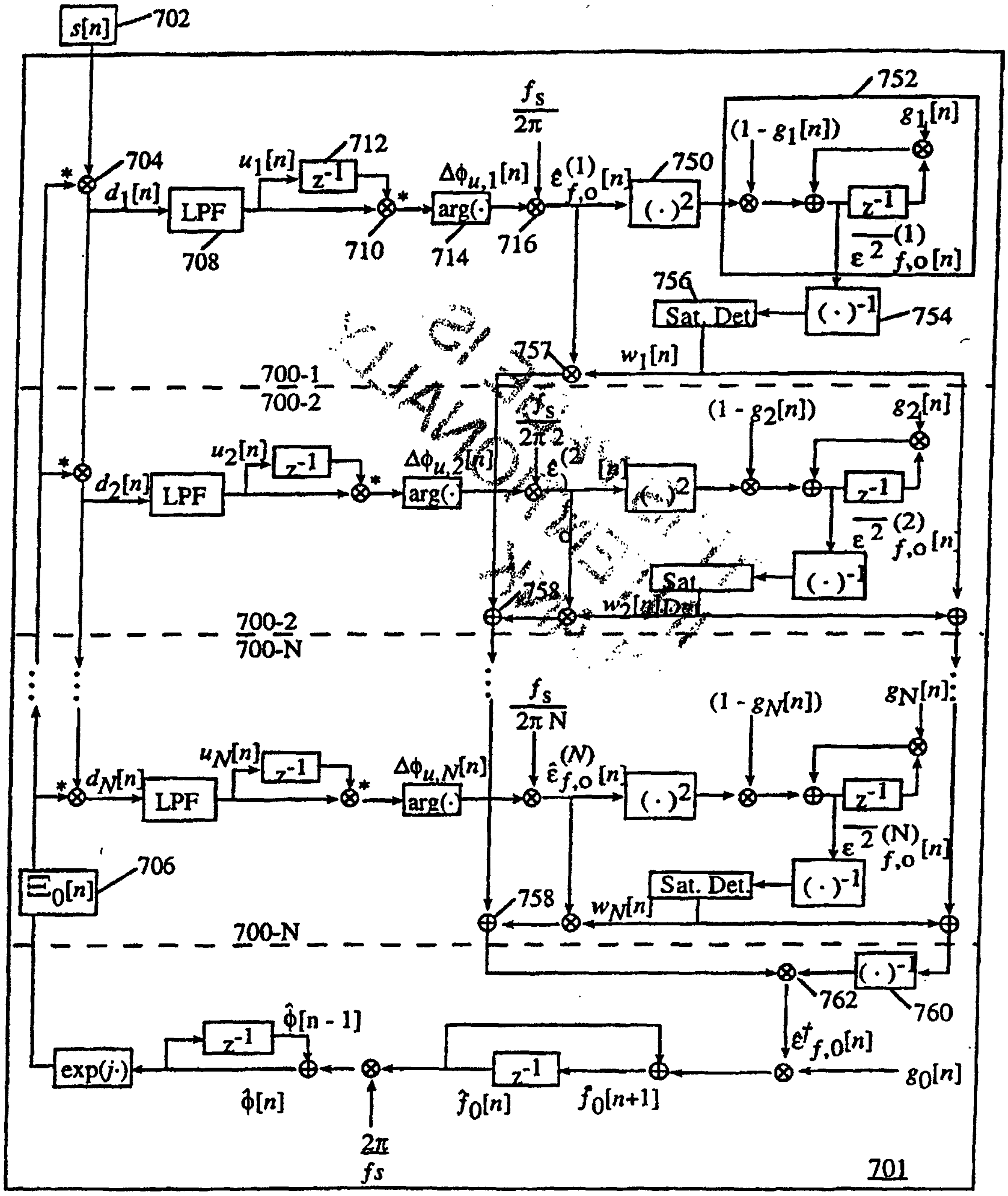

s[n]
702
704
706
708
710
712
714
716
750
752
754
756
757
758
760
762
700-1
700-2
700-N
701
LPF
z-1
arg(·)
(·)2
(·)-1
Sat. Det.
exp(j·)
d1[n]
d2[n]
dN[n]
u1[n]
u2[n]
uN[n]
Δφu,1[n]
Δφu,2[n]
Δφu,N[n]
fs/2π
fs/2π 2
fs/2π N
ε(1) f,o[n]
ε(N) f,o[n]
(1 - g1[n])
(1 - g2[n])
(1 - gN[n])
g1[n]
g2[n]
gN[n]
ε2 (1) f,o[n]
ε2 (2) f,o[n]
ε2 (N) f,o[n]
w1[n]
w2[n]
wN[n]
Ξ0[n]
φ[n - 1]
φ[n]
f0[n]
f0[n+1]
g0[n]
εf,0[n]
2π/fs

FIGURE 7